Intercom Technology 2005

White Paper

Vitec Group Communications

Intercom Technology in 2005/2006

In July 2004, Vitec Group plc, the owners of the Drake 4000, Clear-Com Matrix Plus 3 and Eclipse intercom brands, merged under one Vitec Group Communications banner. This is just the latest chapter in the developments of the former Drake and former Clear-Com matrix products. In this article we explore how the 4000 and Eclipse systems came to being the leading intercom brands and reveal the next developments in intercom for 2005/6.

Intercom is needed where many people work together in some common project, such as making a television programme or directing a live event. For this to happen smoothly, so that responses to questions, commands, and instructions pass speedily between teams, the use of telephones soon becomes impractical. Consider a situation in which the Director is already in a call and blocks another incoming call from the studio Floor Manager to alert the production control room that a guest is not in place, and the Presenter should go to another item. This case of blocking highlights the difference between a telephone system and an intercom system. Unlike a telephone system, the intercom system allows multiple callers to get to a destination, indicates who they are, and provides selective answering.

MATRIX AND PARTY LINE

Two dominant types of intercom systems exist, the party-line and matrix systems. In a party-line system, a common 2-wire cable ring connects together low-cost single-key user stations that conference with each other. The ring connects everyone together efficiently but disallows simultaneous single point-to-point private calls on the same cable. The 2-wire party-line system is prone to noise, and cable interfaces have to be tuned to give the best quality of service.

In a matrix system all users have multiple keys to talk to selected destinations independently. The cable interconnections are based on balanced duplex 4-wires and this makes the cables less prone to interference without any requirement to tune interfaces. Each user has a panel, star cabled to a central matrix, which means that each user can send and receive personalised audio commands. The matrix of inputs to outputs provides a manageable means of providing quite complex communication options that would not be available in a party-line system.

EVOLUTION OF MATRIX INTERCOM

In 1993, the world's first digital matrix intercom, the Drake 3000, was built on the heritage of the analogue systems before it. Drake were early pioneers in taking the "wire-per-crosspoint" hard-wired custom matrices and applying manufacturing standardisation to give broadcast customers the functionality of a custom wired intercom but in a repeatable modular system. Early hard-wired systems in the 1970s and 80s used multi-core wired connections from the user's panel keys to the control switches in a central equipment rack. Each key from every user panel sent an earth-free 0-volt switching contact to close an input audio gate to a destination audio driver. Equipment racks of multi-channel circuit cards provided input buffering, level control, switching, crosspoint logic, output audio and voltage distribution. The analogue Drake 600 series successfully modularised these functions so that customers could tailor their complex intercom requirements into a set of standard cards and frames. These wired systems were characterised by a lack of a single fixed audio distribution bus and relied on crosspoint cards mixing to multiple virtual earth connections on the output audio cards.

Changes to these hard-wired systems were very cumbersome and required delicate wiring changes at the back of inter-linked card frames. The whole game changed with the introduction of the PC to the broadcast intercom industry. In 1987 Drake were the first to provide a PC GUI (Central Configuration Facility) to allow users to quickly define and edit a map, or crosspoint database, without resorting to point-to-point wiring. The newly developed Drake 6000 series used a Motorola 68000 Central Processor unit which now defined the routing between synchronised inputs and output audio bus time slots through a 16-channel multiplexed routing switch card. The panel key control system now used RS422 serial data format from the user panels, and dual CPUs provided a level of redundancy not seen in the earlier hard-wired systems. In both the Drake 6000, and later in the Clear-Com Matrix Plus, for the first time intelligent user panels had CPU and memory, giving updateable displays allowing the re-labelling of keys from the central configuration program.

THE DIGITISATION OF INTERCOM

From the 6000 digitally controlled analogue routing system, Drake Electronics Ltd developed the 3000 system which was the first in the world in 1992 to offer a truly digital Intercom. In 3000, the 16-bit digitised intercom audio was routed digitally from input source to output destination busses through a four-way fast digital audio router, the FRM. The card had a MDAC to provide digitised audio gain and the use of four routers allowed very short through times needed to accomplish all system routing in less than 35 ms. Every "crosspoint" was now an instruction to route input sample(s) from a TDM source bus to a TDM slot in the output bus with a gain setting coefficients.

The digitisation of audio into addressable digital audio samples made a considerable difference to the established analogue routing found, for example, in the Drake 6000, which it replaced. For the first time a 3000 series

digital 128 x128 port matrix was available in just 9 rack units, while previously such a matrix had required at least one equipment bay. Also the separation of the routing function from the input and output gave for the first time the possibility of backing up the routing to give crosspoint redundancy. This was in addition to making the control system redundant, an industry practice even then.

The Drake 3000 series extended its claim to being “truly digital” in first providing user panel connections to the central matrix over co-axial cable.

Diagram: Clear-Com Eclipse matrix – the newest and most advanced digital matrix on the market today

The multi-core cables in CPU-driven analogue systems needed at least four pairs to get the serial data and analogue audio to the main frame. In the 3000 series matrix, digitised audio between the matrix and panels used a bi-directional stereo AES/EBU format with key and display control data embedded in the user bits. This provided a quick connection using industry standard 75-ohm video cable and gave the additional benefit of two audio mixes back to the user panel in one cable. A panel user could have a separate mix of intercom from cue programme but use just one co-ax cable.

NETWORKED INTERCOM SYSTEMS

During these advances in intercom digitisation there were also changes in production techniques throughout the broadcasting world. From single studios handling all in-house and incoming remote communications came larger productions with multiple outside studios intelligently networked to provide for large sports or election-type programmes. At the same time developments in telecom technology allowed audio and data connections to be managed through the public switched networks. By the time the 3000 series had evolved into the early 4000 series 1 (1998) provision had been made to provide connections to ISDN and soon after, E1.

In a networked intercom system, a user can make calls out of the local matrix through a distant remote matrix to another user. The routing not only of audio, but also of control data, provides users with both call signalisation and key

labels needed to identify incoming calls. In intelligently linked systems, like the Drake 3000 and 4000 series and the Eclipse, this routing is managed through a central PC application setting inter-matrix connections by audio trunks and linking the PC and matrices over a common data LAN. The trunk lines transport individual "conversations" between sites. A local Director could send production direction commands over a single audio trunk line to many users in a remote system. The use of trunk lines to take pre-mixed audio between one to many users provides a useful cost efficiency that allows telecom lines to be used.

Until the advent of ISDN in the mid '90s all remote trunking was done through 2-wire conversion of the audio over PSTN and either separate serial data by modem or by multiplexing audio over data through single high-speed modems. The latter operation worked well if bandwidth was available. The advent of dual 64-Kbps ISDN systems offered reliable dial-up communications with both audio and data in single 64-Kbps streams. The Drake VeNiX system, which is offered with the 3000 and 4000 systems, provides two intelligent trunk lines per ISDN-2 service, one per bearer channel. The VeNiX ISDN CODEC uses fast G.722 7-KHz audio compression and serial data multiplexed into a single 64-KB ISDN bearer. This gives a dial-up bi-directional trunk line.

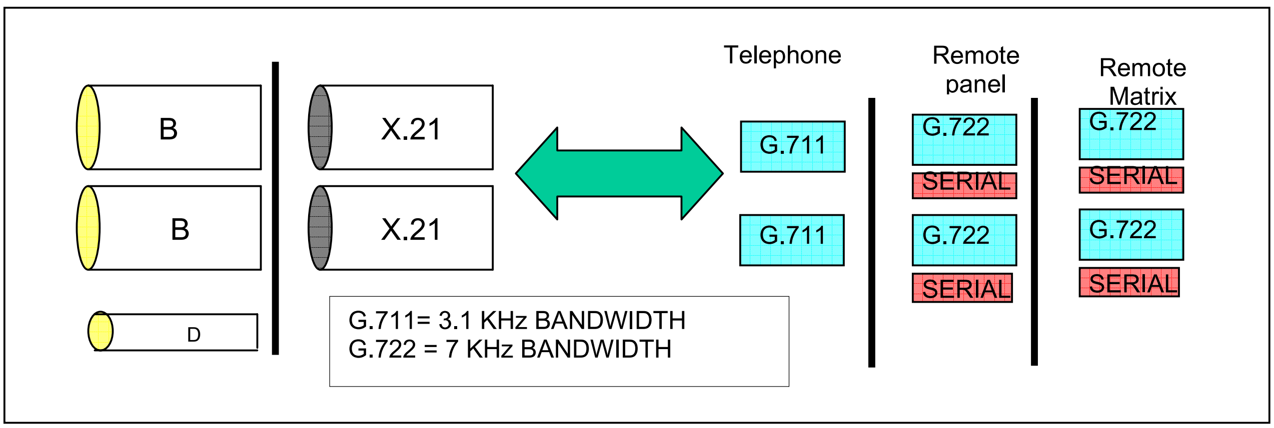

Diagram: VeNiX data architecture or network

The central matrix processor switches the ISDN line according to the remote trunk requirement, auto-dialling the destination VeNiX CODEC as required. ISDN dialling is very fast at less than 2 seconds to establish a line. The 4000 system code can hold the line open for a configurable amount of time after the last trunk request, giving near instantaneous remote communications on demand. The dial-up ISDN telecom trunk lines save costs on any nailed up digital system providing a very reliable, efficient, and high quality of service link. Configuration through the Windows© GUI in CMAPSi is simply a table of potential links with telephone numbers that the system dials on remote calls.

Conferences, or internal matrix party lines, have been used since the earlier days to reduce the need to make multiple manual connections when all users need is an "all call" key to talk and listen to a non-predefined group. The conference is likened to a "software" bus that users can key to talk and listen

to with the proviso that they hear everyone else but themselves. With the move to extended TV productions, networking the conferences between systems is essential. The ISDN dial-up conference operation is managed by a central matrix dialling up the others for a system-wide conference on demand.

Although ISDN provides a cost effective trunking system for low-to-medium traffic between remote systems it becomes less cost effective for high traffic connectivity. The 4000 series II provides a 30-channel E1 connection when the traffic demands many connections between systems, either over a telecomm link or directly. The Drake 4000 Hi-Que and Eclipse E-Que cards provide dual 30 x G.722 audio trunk lines per card. The E1 signal is a 2-Mbps 32 x 64KB data stream. Again this uses G.722 fast audio compression, and in its simplest form provides 30 trunks over a single CAT5 cable. The E1 is a widely available telecomm digital standard that is often used between PABX switches. The wide availability, low latency, and plentiful third-party device support makes the Hi-Que E1 card the connection choice in many site-to-site networks.

Redundancy is important considering that all 30 channels can be lost when a single E1 is removed. With the 4000 and Eclipse E1 cards, the second E1 port can be made to mirror the primary E1 port. In the event that the primary E1 link is broken, the secondary E1 link takes all the routes between the same destinations. The diagnostic systems will then report such failure. The E1 system carries a synchronising signal that can be used for failure recognition.

IP NETWORKING

One such third-party operation with E1 is to provide TDM audio over IP. IP communication offers the potential for low-cost wide communications. In practice the widely available Voice over IP provides packetised digital audio and data over an enterprise WAN or Internet.

The Clear-Com **VoICE** VoIP interface provides up to four channels of panel-to-matrix and/or matrix-trunk to matrix-trunk communication over IP. In well-managed and controlled bandwidth, VoIP can offer very acceptable results. Although in most cases VoIP suffers from IP packet loss, jitter and latency, with careful design these can be much reduced by modern techniques. Packet losses through queuing in routers can be kept to a minimum but losses cannot usually be regained without adding delay. It is not unusual to have packet losses of 30 ms per 5 seconds. The VoICE interface uses type-of-service control bits to help lower packet losses and jitter by providing audio priority in the network. If latency, the delay in transmission, increases beyond 50 ms one way, intelligibility can be prevented, especially when many IP audio paths are mixed into conferences. VoICE uses real-time protocol with type-of-service bits in each channel to minimise the latency in real networks. VoICE also uses digital echo cancellation technology to dynamically configure the high bandwidth duplex routes to offer real improvements over conventional communication over IP. VoICE can be used comfortably for all non-critical intercom. This can include studio centre to remote bureau, inter-area intercoms, remote operational centres and off-line production centres.

Where more critical communication is required, and IP links exist, a predictably more stable and lower latency operation than VoIP is TDM over IP. In this multi-channel trunk operation the 30 E1 audio/data channels from the Clear-Com matrix are parallel processed with a single IP overhead into TDMoIP streams. The jitter between adjacent audio channels is much lower, the bandwidth at G.722 coding is twice VoIP toll quality, and the delays are lower and predicable. It has been demonstrated that TDMoIP can offer much better audio quality for remote conference working and is fast becoming the choice for low-cost multi-channel trunk operations. Telecom E1 standard IP muxes offer up to 4 E1 ports per IP stream, and as VoICE interfaces also bridge the Ethernet LAN control data over the same system. This solution provides the customer with a low-cost centre-to-centre communication system where many users can communicate over IP trunks with low and predictable delay that enables conferencing to be achieved.

The use of Telecom standards offers intercom manufacturers such as Vitec Group Communications easier connection at lower costs through common standards and wide IC support than by making proprietary solutions. However, in the field of highly available interconnections where the matrix frames are not required to operate over telecom lines and many matrix-to-matrix links are required, single high-bandwidth fibre links prevail.

FIBRE SYSTEMS

The fibre connection of matrix audio busses provides the possibility to link multiple matrix frame audio back plane busses into one configurable fibre bus. For the Vitec Group Communications Eclipse and 4000 matrices, the Fibre-Net system offers more than 1300 fibre audio time slots throughout the matrix network which may encompass up to 127 matrix frames. Simple single or multi-mode fibre links can be duplicated into a countra-rotating dual-concentric fibre connection offering full redundancy and self-repairing highly available connectivity.

One of the advantages to the trunk systems when compared to highly connective fibre systems is that they are efficient in processor overheads, easily managing the relatively few interconnections. Fibre connectivity demands a great deal more from the central processing and for this reason high-specification leading-edge processing is now required to provide reliable fibre connectivity without having the systems real-time resources collapse. The Fibre cards for the Eclipse and 4000 use highly capable processing to provide both control and audio facilities, such as pre-mixing with level control, into the fibre network. Using the strengths of the 4000 trunking in combination with the Fibre bussing allows further savings in processor overhead by premixing multiple audio inputs into a smaller number of timeslots. This can be used when a remote panel user wants to listen to a collection of audio ports at one listening station. The user will be able to make level adjustments to the mix and these will be sent as gain coefficients to the FPGAs on the Fibre bus cards.

Fibre connectivity requires similar redundancy to that provided with the Hi-Que E1 trunks. In the 4000 and Eclipse Fibre-Net systems, the Fibre cards have dual-fibre transceivers which can be wired to fibre pairs in contra-rotating busses. In this way the fibre rings can be segmented in failure modes and still perform. Here are some redundancy examples:

- If a single fibre connection is lost on the ring, the matrix nodes adjacent to the failure will loop-back their connections to the failed cable, healing the ring.

- If a matrix node is lost on the ring due to a fibre-optic linking card failure, the nodes adjacent to the failed node will loop-back their connections to the failed node, healing the ring.

- If two adjacent fibre connections are lost on the ring, this will be handled as for the loss of a single node, where the nodes adjacent to the failed node will loop-back their connections to the failed node, healing the ring.

- If two non-adjacent fibre connections are lost on the ring, the nodes adjacent to the failures will loop-back their connections to the failed cables, healing the ring into two separate smaller rings.

- If two adjacent nodes are lost on the ring, this will be handled as for the loss of a single node where the nodes adjacent to the failed node will loop-back their connections to the failed nodes, healing the ring.

The possibility to "spur" a fibre ring from any one of the matrix nodes in order to add a new matrix into the network—for example, to add an outside broadcast vehicle into the studio system without breaking the network, is an important feature of the Fibre-Net system. In this case, one matrix operates as a gateway by hanging on both rings at the same time. The Eclipse matrix, for example, will then seamlessly cross-connect the audio from one ring to the other as required.

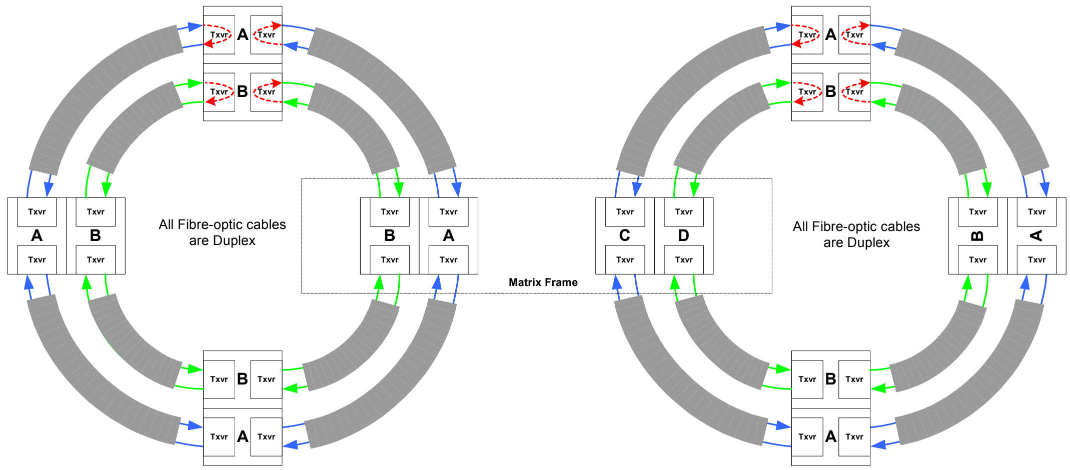

Diagram: 4000 and Eclipse Fibre system architecture showing dual concentric rings and "spurring" to another ring.

SUPERVISOR CONTROLS

In these large communication systems, perhaps providing an inter-area intercom throughout a studio complex or over multiple sites, there is a requirement that managing the connections effectively is relatively easy. The use of on-line supervisory diagnostics and monitoring activities helps to provide for central support. The 4000 system provides a Supervisor panel operation that allows a target users LCD key panel to copy its control surface to the Supervisor panel together with the matrix which copies all audio to the target to the Supervisor panel also. In this way the supervisor can make adjustment through the LCD key panel's audio local assignment and mix controls for the target user, at the same time as listening to the targets audio mix.

Diagram: Refresh LCD key panel – supervisor or target panel

DIGITAL WIRELESS

Of course it would be very useful if all of these "panel facilities", such as labelled keys, local access to all ports, and call signalisation, were available in a wireless system. With FreeSpeak™, the Clear-Com and Drake Digital Wireless system, they are. The FreeSpeak system provides full-duplex digital audio and data over a licence-free DECT link from active antennas from the matrix through E1 distribution. The DECT radio space is divided into 240 timeslots in the band 1881.792 to 1897.344 MHz. Each audio route uses two timeslots, four for a full send-and-return duplex communication. This gives 60 channels divided across 10 active antennas. Each active antenna handles five beltpack attachments plus an extra make-before-break hand-over channel, six channels each.

The unique feature of FreeSpeak is that beltpack users can freely roam across all available antennas so that for the first time a wired panel user can key directly to a wireless beltpack by name and the system follows that beltpack no matter where it is. This is a Vitec Group Communications patented feature called "Dynamic Port Allocation".

CENTRAL CONFIGURATION SOFTWARE

The central configuration of these key elements in a modern intercom system —panel key assignments, port grouping into fixed or dynamic groups,

conferences/party lines, and networking through trunks and/or fibre—is vital to gaining acceptance with the end user. The configuration GUI must give the non-frequent user the confidence to understand, explore, and make changes whilst the system is in use. Vitec Group Communications have released the ECS software platform for both Eclipse and soon 4000 users, which provides user familiarity through a Windows© XP explorer look and feel. The ECS software provides graphical panel assignment that lends itself to intuitive click and drop mouse operation and allows the panel's controls to be driven from the PC screen for centralised supervisory control.

Diagnostics are very important in potentially large fibre or trunked networks. ECS provides full status and error reporting through to the logging PC server. Client PCs, logged on to the central server, can run history event logs or search for particular events from the central database. To aid remote diagnosis, a web client sits on top of the server and allows users to gain access to the event and status logs over the Internet. The central server connects to the 4000 (with the latest Processor Cards) or Eclipse systems over a Dual Ethernet LAN for redundancy in the control sub-systems.

The VGC 24x7 customer support operation makes good use of these remote diagnostics, providing another level of customer confidence and service beyond the office hours telephone help line.

The ECS software design lends itself to adding modules for specific tasks. The existing 4000 Master Control Room lines management module, MCR, will migrate into the ECS system offering fast assignment of 4-wire audio, AES audio, and ISDN lines to remote panels completely online.

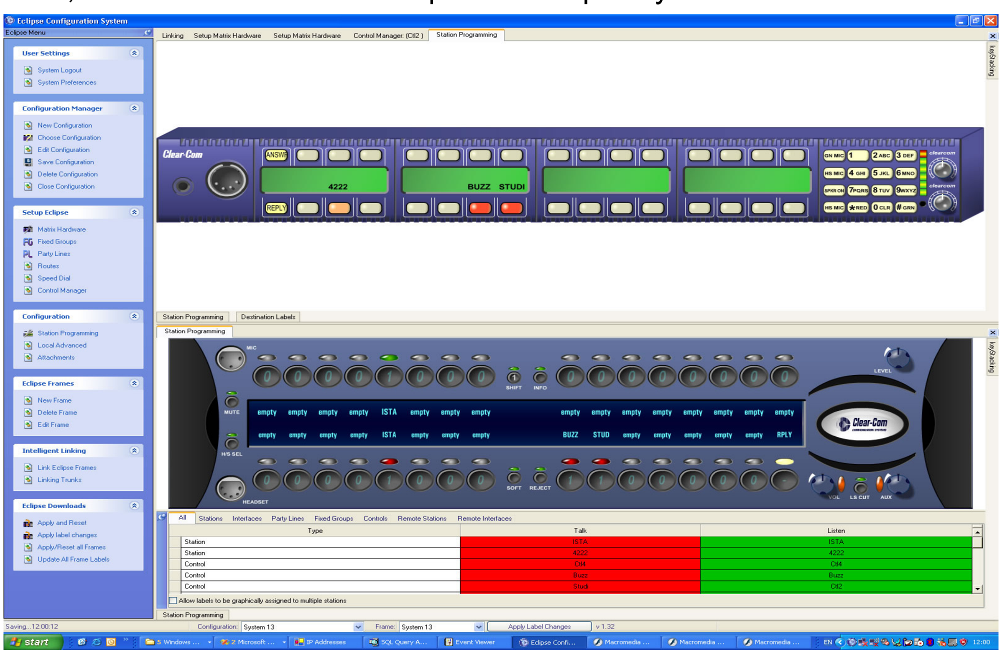

Diagram: ECS Panel key assignment page showing click and drop key and on-line PC supervisor capability

The ability to modify sub-systems within an intercom system has made the Drake 4000 very compliant with customers' requests for targeted solutions. The evolution of this customisation led to the Gemini 4000 product for Air Traffic Control, now a successful branched development at VGC.

Today's requirements for intercom have come a long way in 20 years. Modern systems such as the Eclipse and 4000 provide users with immediate response to system changes, full diagnostics, low-cost routing across telecom systems, convergence with Ethernet LAN systems, digital wireless mobility, and smaller nodes sitting within a fibre network, but all sharing the total system's resources. With all the changes in the media and arts we can expect only the most flexible systems with these high intra-system and external inter-connectivity options to survive.

Simon Browne, Matrix Product Manager
Vitec Group Communications
Cambridge Research Park
Cambridge, United Kingdom